

ABTO RTP SDK

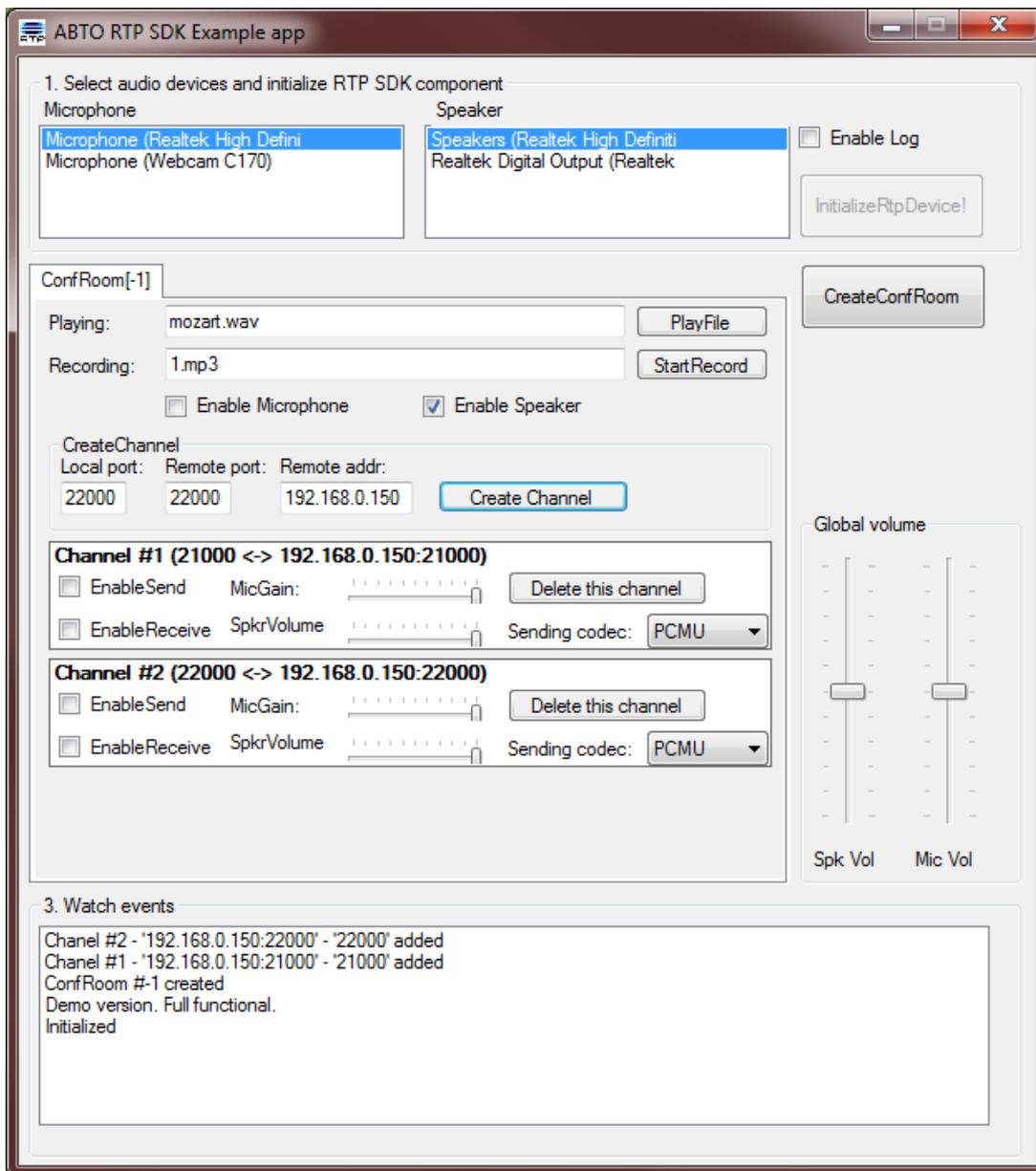
Overview

RTP SDK provides a powerful and highly customizable solution to quickly build application with ability send/receive sound by RTP protocol.

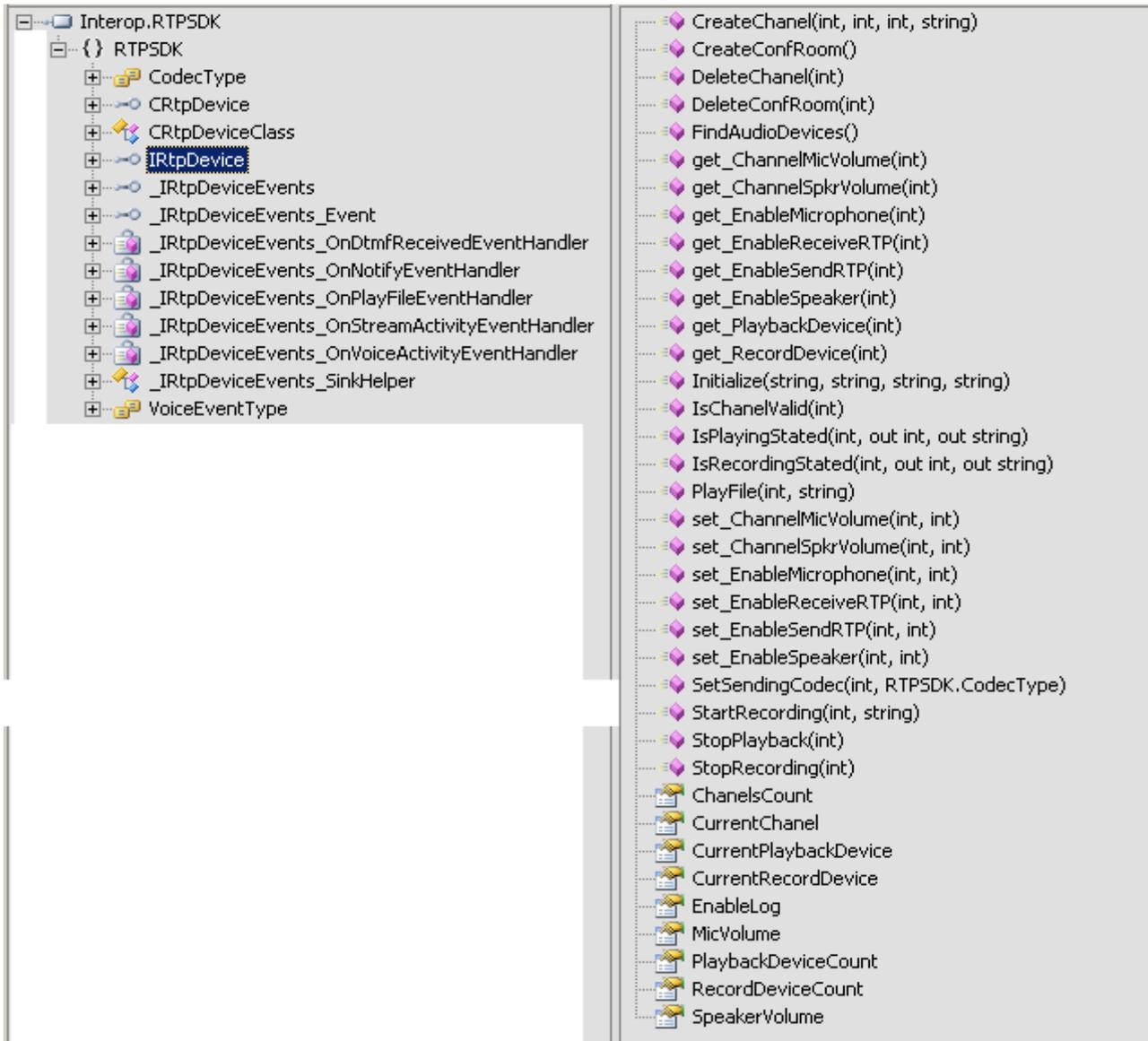
Main features:

- Ability to create multiple independent conference rooms
- Ability to create multiple channels in each conference room
- Ability to select different sending codec for each channel (G711, G726, G729, GSM, SPEEX)
- Recording in mp3 format
- Playing wav, mp3 files
- Voice detection feature

GUI of the RTP SDK example application:



RTP SDK component has following methods/properties/events:



Methods	Description
CreateConfRoom	<p>Creates conference room. Returns id of the created room as integer.</p> <p>Each room has own mixer. Example: We have created conf.rom with 3 channels (A,B,C) and enabled microphone and speaker (Mic,Spkr). When channel A has enabled sending SDK will send to this channel voice from channel B and C (when receiving enabled) and voice from Mic (when enabled). It can be shown as: A = B+C+Mic B = A+C+Mic C = A+B+Mic Spkr = A+B+C</p>
CreateChanel	<p>Creates channel in conference room.</p> <p>Input arguments: LONG ConfRoomId, LONG LocalPort, LONG RemotePort, String RemoteAddr</p> <p>Returns id of created channel.</p>

DeleteChanel	<p>Deletes previously created channel by its id.</p> <p>Input argument: LONG ChannelId</p>
DeleteConfRoom	<p>Deletes previously created conf.room by its id.</p> <p>Input argument: (LONG confRoomId)</p>
FindAudioDevices	<p>Finds existing audio devices in system. Has to be invoked before <code>Initialize</code>, and provide ability for user selected audio devices.</p> <p>Note: When audio devices are not connected or not selected SDK can work without them, all local audio features will be disabled.</p>
Initialize	<p>Initializes component</p> <p>Input arguments: String PlaybackDevice String RecordDevice - selected playback (record) device.</p> <p>String LicenseUserId String LicenseKey - license data.</p>
PlayFile	<p>Start/stop playing file for selected conf.room.</p>
StopPlayback	<p>Input arguments: confRoomId –conf.room id, received from method <code>CreateConfRoom</code>.</p> <p>Note:</p> <ul style="list-style-type: none"> A. Method plays sound for all channels in the conf.room. B. When input file is in mp3 format SDK can automatically convert it to wav, play, remove after playing.
IsPlayingStated	<p>Returns true when playing has already started on selected conf.room, and also returns path to file which is playing.</p> <p>Input arguments: LONG ConfRoomId, out string FilePath</p>
StartRecording	<p>Starts/stops recording file for selected conf.room. SDK records mixed sound from all channels and local microphone, compresses and stores in mp3 format.</p>
StopRecording	<p>Input arguments: confRoomId –conf.room id, received from method <code>CreateConfRoom</code>.</p>
IsRecordingStated	<p>Returns true when recording has already started on selected conf.room, and also returns path to file which is recording.</p> <p>Input arguments: LONG ConfRoomId, out string FilePath</p>
SetSendingCodec	<p>Set codec that will be used for sending audio data.</p> <p>Input arguments: LONG ChanellId, CodecType codec Possible values for 'CodecType' are: eG711_mulaw, eG711_alaw, eGSM, eILBC, eG729A, eSPEEX, eG726</p>

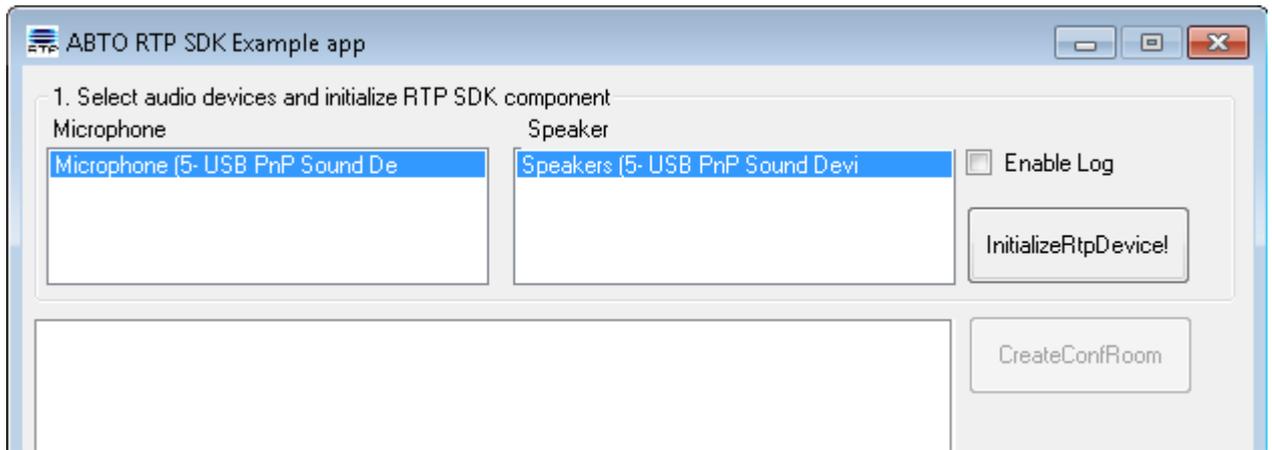
	<p>Note: To apply new codec is required to stop and start sending again.</p>
<p>set_ChannelMicVolume get_ChannelMicVolume</p>	<p>This property allows to set input volume of the conf.room microphone which will be sent through this channel.</p> <p>When property set to 0 – channel will not send sound from microphone.</p>
<p>set_ChannelSpkrVolume get_ChannelSpkrVolume</p>	<p>This property allows set volume of the received voice before mix it and end to speaker.</p> <p>When property set to 0 – speaker will not play sound from this channel.</p>
<p>set_EnableSpeaker get_EnableSpeaker</p>	<p>Enables/Disables speaker for selected conf.room.</p>
<p>set_EnableMicrophone get_EnableMicrophone</p>	<p>Enables/Disables microphone for selected conf.room.</p>
<p>set_EnableReceiveRTP get_EnableReceiveRTP</p>	<p>Enables/Disables receiving RTP on selected channel.</p>
<p>set_EnableSendRTP get_EnableSendRTP</p>	<p>Enables/Disables sending RTP for selected channel.</p>
<p>SpeakerVolume</p>	<p>Changes global speaker volume. Input arguments: LONG Level – level in range [0..100]</p>
<p>MicVolume</p>	<p>Changes global microphone gain. Input arguments: LONG Level – level in range [0..100]</p>
<p>CurrentPlaybackDevice PlaybackDeviceCount PlaybackDevice CurrentRecordDevice RecordDeviceCount RecordDevice</p>	<p>Methods that returns found playback/record devices</p>
<p>EnableLog</p>	<p>Enables/Disables internal log features. When enabled - SDK writes file 'RtpSdkLog.txt' in same where component is registered. Note: this property has to be invoked before 'Initialize'.</p> <p>Input arguments: LONG Enabled - 1-to enable log, 0 – disable.</p>
<p>Events</p>	
<p>OnNotify</p>	<p>SDK generates this event when it has some additional information about device state. Event arguments: String Msg</p> <p>Example: When started file playing SDK generates this event with message: <code>OnNotify("Playing started: c:\mozart.wav")</code></p>

OnStreamActivity	<p>SDK generates this event when when detected activity on some channel.</p> <p>Event arguments: String Msg String Addr LONG port</p> <p>Example: When stream started: <code>OnStreamActivity("STREAM_START", "192.168.0.56", 21000)</code></p>
OnPlayFile	<p>SDK generates this event when finished playing sound file.</p> <p>Event arguments: VoiceEventType eventType - always equal 'eStopped' LONG ChanelId - 1-based channel id where playing finished.</p>
OnVoiceActivity	<p>SDK generates this event when recognized some voice activity in received data. This event can be used for start/stop recording voice without silence.</p> <p>Event arguments: VoiceEventType eventType - equal 'eStarted' when voice started and 'eStopped' when stopped. LONG ChanelId - 1-based channel id.</p>

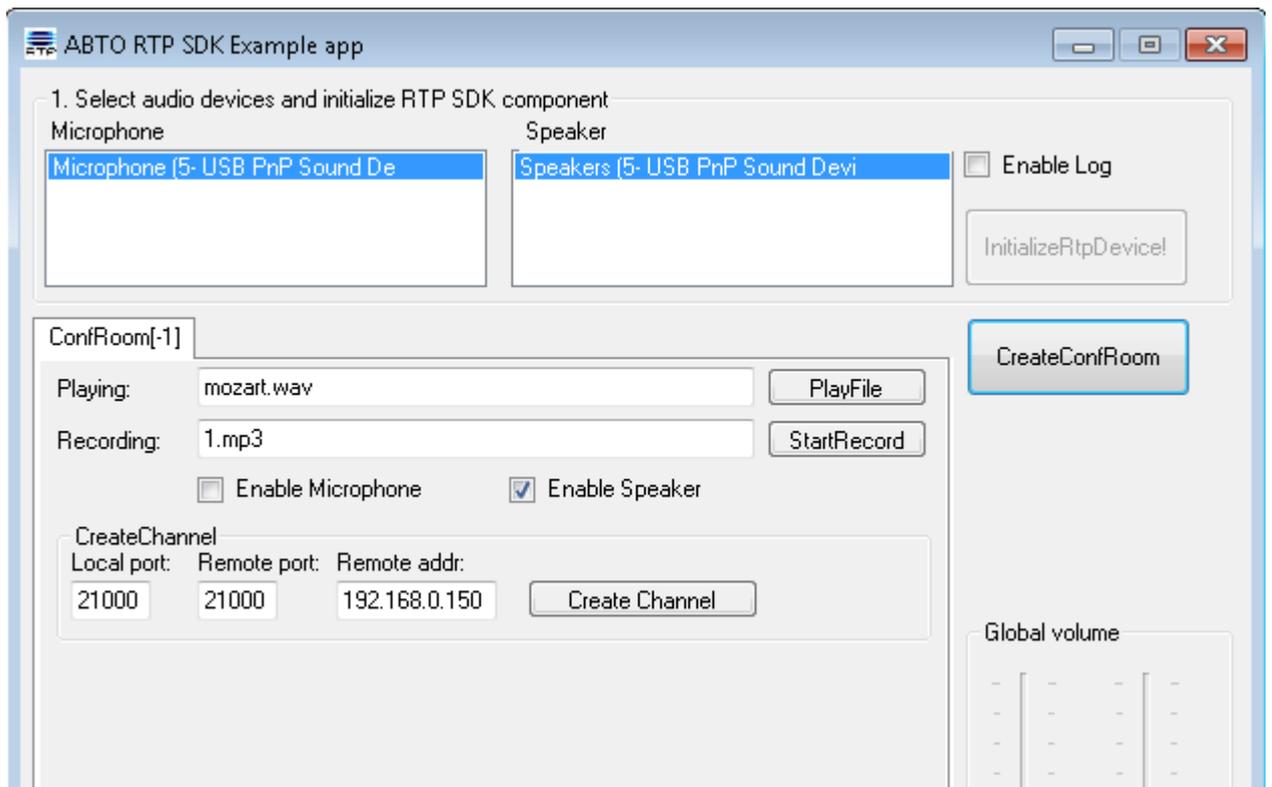
How to use example application

See source code of the example application <install_dir>\SampleWindowCS\SampleWindowCS.sln

1. Select audio devices and click "Initialize RTP device"



2. Create conf.room by clicking "CreateConfRoom" button.
SDK/application allows create multiple conference rooms, for each room user can start playing, start recording, enable speaker and microphone. Room mixes sound from all channels as explained above in method "CreateConfRoom".



3. Create channels.

Input local port, remote port and remote address and click "Create Channel".

Each channel independently allows enable sending/receiving.

